IJIREEICE

ISSN (Online) 2321-2004
ISSN (Print) 2321-5526

International Journal of Innovative Research in
Electrical, Electronics, Instrumentation and Control Engineering

Vol. 8, Issue 1, January 2020

Using FIR Coefficients to Form a Voiceprint

Dr. Amjad Hindi[1], Prof. Ziad Alqadi[2], Dr. Ghazi M. Qaryouti[3],

Prof. Mohammad Abuzalata[4], Prof. Yousif Eltous[5]

Albalqa Applied University, Faculty of Engineering Technology, Jordan, Amman[1-5]

Abstract: Voice recognition is a method of security that can be accessed almost by everyone. It provides those with visual impairments the same level of security as those without. Similarly, technologies that require physical interaction may prove difficult barrier that other methods of technology may carry. Voice recognition systems are typically utilised in contact centres where there is no other method for physical authentication. However, it can be used across a wide range of industries such as: Hospitality**,** Retail and Logistics**,** and Banking and Finance**.** In this research paper we will introduce a method of creating human voiceprints based on using finite impulse response filter coefficients, this linear prediction coding method will be tested, implemented using various wave signal files to prove the act of how to identify the person and how to identify the spoken word or phrase, allowing the recognition system to use a voiceprint as a password.

Keywords: Digital Signal, Voice Signal, Voiceprint, FIR, LPC, Filter Order Recognition System, Correlation, MSE

I. INTRODUCTION

Digital signals [1], [2], [3] such as color digital images [4], [5], [6], digital voices [7], [8] are the most papular digital data types used by various applications [9], [10] such as security systems [19], identification system[20], [21] and human recognition systems [11], [12].
Information in the voice signal is embedded in both its time and frequency domains, within these domains; information relevant to profiling may be present in the patterns exhibited by specific characteristics of the voice signal. The signal may however also have characteristics that are not evident in these domains, and must be searched for in other (derivative) mathematical domains where the relevant patterns become more tangible for measurement and analysis. This, however, is the subject of feature discovery—a subject that is discussed in FIR modelling. A third domain that reflects the information in the voice signal is that of physical or abstract models that simulate or explain the voice signal and the processes that generate it. We will refer to this as the model domain [18], [20].
Sound signal is a continuous wave [13-17]; it is an analog signal. This means that one cannot detect the precise moment the pitch changes. Capturing this continuous wave in its entirety requires an analog recording system; what the microphone receives is exactly what's written onto the vinyl disk or cassette. Analog is believed to be the true representation of the sound at the moment it was recorded.
Digital sound is not a recording of the actual sound, but rather a combination of binary code, the utmost simplest machine language of zeros and ones, representing the sound's intensity and pitch at precise intervals with relative accuracy. The binary code is arranged in a specific pattern informing the computer how to recreate the sound itself. It is not a single wave the way analog sound is, but rather a composite of multiple segments representing consecutive moments of intensity and pitch. Where an analog recording is similar to the fluency of film, a digital recording is stop motion photography.
Analogue voice signal can be converted to digital (discrete) signal by applying sampling and quantization as shown in figure 1. The digital signal then can be manipulated to create some features, which can be used as a voiceprint to recognize the person and/or recognize the spoken words.
Voice recognition works by analyzing physical and behavioral factors to produce a unique voiceprint for each individual. These factors include pronunciation, emphasis, speech speed and accents, and also physical characteristics such as vocal tract, mouth and nasal passages (see figure 2) [22], [23] .
Voice identification occurs in two forms: It can be passive, where a user’s speech is matched to a previously recorded voiceprint. Or it can be active, where callers are asked to recite a predetermined passphrase and their voice matched to the voiceprint saved in a database [24].
A voiceprint is created by reducing each spoken word to segments composed of several dominant frequencies that are then segmented by tones captured in a digital format. Those tones collectively identify the speaker's unique voice print which becomes their unique vocal pattern [25], [26], [27].
Voice identification has common business uses within contact centres, visitor management and in consumer applications for access [28], [29], [30].

Voice recognition systems are typically utilised in contact centres where there is no other method for physical authentication. However, it can be used across a wide range of industries such as: Hospitality**,** Retail and Logistics**,** and Banking and Finance**.**

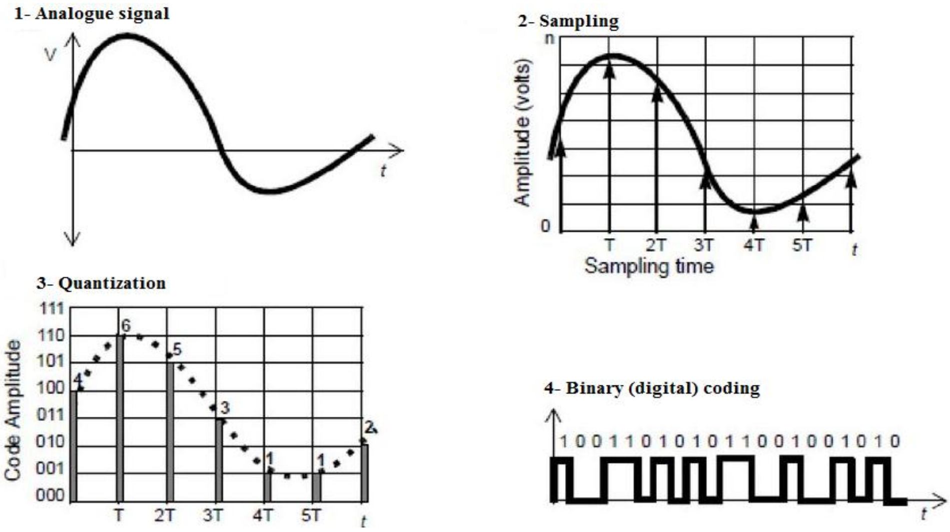

Figure 1: Converting analogue signal to digital signal

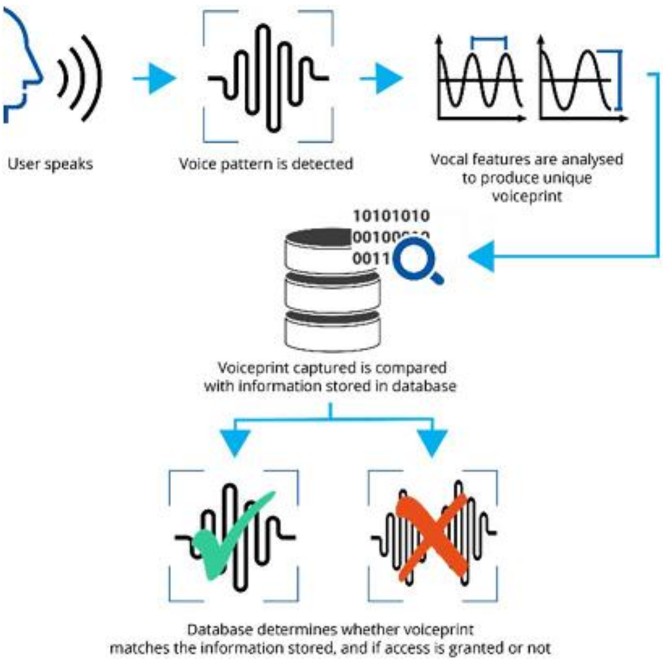

Figure 2: Voice recognition steps

II. DIGITAL VOICE MODELLING

In this research paper we will focus on the model domain to create some values, which are used to form a voiceprint which can be used in a recognition system to identify the person and the spoken word.
Using finite impulse response (FIR) filter we can model the voice signal using equation 1:

$$y(n) = \sum_{k=0}^{M-1} b_k x(n-k) \qquad (1)$$

For 4 order FIR filter equation 1 can be represented using equation 2 [20]:

IJIREEICE

ISSN (Online) 2321-2004
ISSN (Print) 2321-5526

International Journal of Innovative Research in
Electrical, Electronics, Instrumentation and Control Engineering

Vol. 8, Issue 1, January 2020

$$y(n) = b_0x(n) + b_1x(n-1) + b_2x(n-2) + b_3x(n-3) + b_4x(n-4) \qquad (2)$$

The term finite impulse response arises because the filter output is computed as a weighted, finite term sum, of past, present, and perhaps future values of the filter input, i.e., Figure 3 shows the 4 order filter diagram

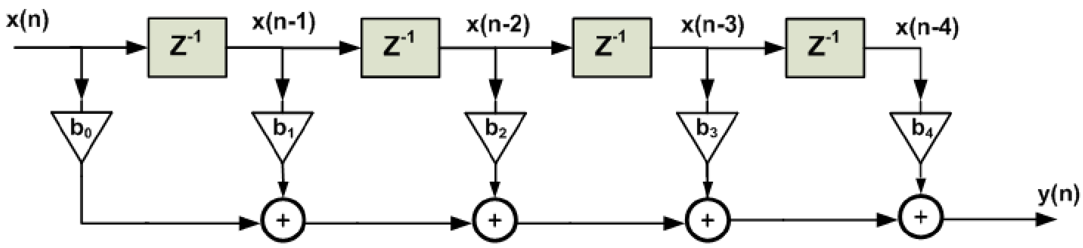

Figure 3: 4 order FIR filter diagram

Using LPC matlab function we can determine FIR filter coefficients for any filter order, these coefficients can be used later by a filter function with z-transform shown in equation 3 to generate the estimated voice signal:

$$Y(z) = \frac{0 \;+ b(2)z^{-1} + \ldots + b(nb+1)z^{-nb}}{1}X(z) \qquad (3)$$

The process of FIR filter coefficients calculation and using these coefficients to generate the estimated voice signal is shown in figure

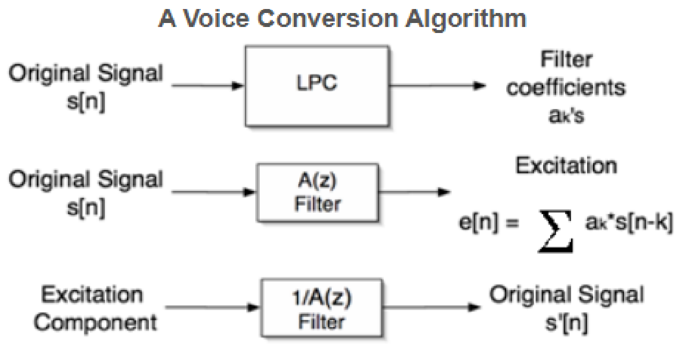

Figure 4: Generating the estimated voice signal

Figure 5 shows the original voice file and the estimated one after applying matlab LPC and filter function using 4 order FIR filter, while figure 6 shows 100 samples of the 2 files.

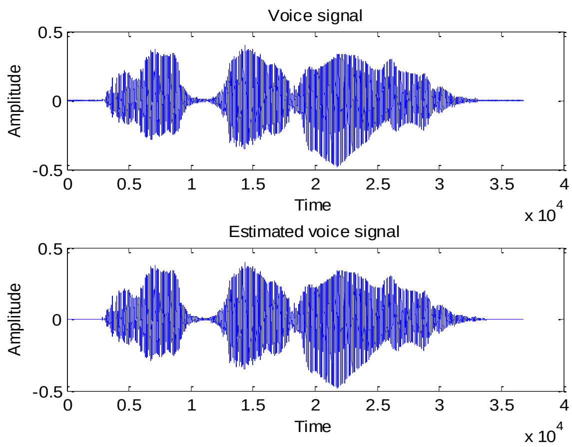

Figure 5: Original and estimated voice files

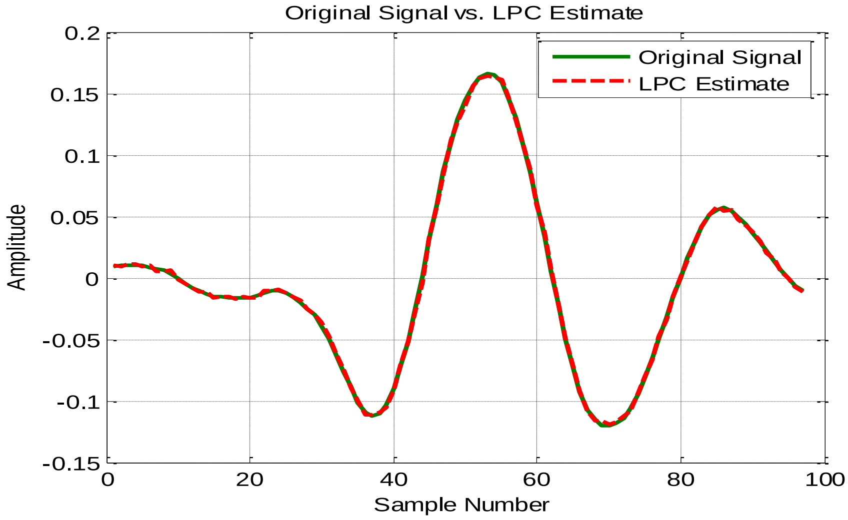

Figure 6: 100 samples of the original and estimated voice files.

Table 1 shows the calculated FIR coefficients and the mean square error and correlation between the original and the estimated voice files using 6 voice samples for the same word spoken by 6 different persons, we can see from this table that the original and the estimated voice files are very closed each to other.

Table 1: FIR coefficients and correlation and MSE between original voices File and estimated one

Voice signal	**FIR coefficients**					Correlation	MSE
Bye-bye-1	1.0000	**-2.8293**	**2.9235**	**-1.2684**	**0.1879**	0.9997	0.0000058575
Bye-bye-2	1.0000	**-2.8967**	**3.2323**	**-1.6911**	**0.3700**	0.9996	0.0000054909
Bye-bye-3	1.0000	**-2.1305**	**1.8298**	**-1.0827**	**0.4333**	0.9970	0.000016928
Bye-bye-4	1.0000	**-1.8329**	**1.4566**	**-1.1583**	**0.6069**	0.9934	0.000063580
Bye-bye-5	1.0000	**-2.5383**	**2.6340**	**-1.4634**	**0.4105**	0.9971	0.000030758
Bye-bye-6	1.0000	**-1.6374**	**0.6464**	**-0.1703**	**0.1986**	0.9958	0.000013331

III. IMPLEMENTATION AND EXPERIMENTAL RESULTS

Various methods are used to extract voice signal features, these methods were described in [30], [31], [32], [33], [34], [35], [36], here we will introduce LPC method by applying matlab lpc function using various voice signals.
Experiment 1: Using 4 order FIR to form voice features for the same word spoken by 6 different persons:
Table 2 shows the results of this experiment:

Table 2 : Experiment 1 results

Voice signal	**Features(Original voice signal)**			
Bye-bye-1	**-2.8293**	**2.9235**	**-1.2684**	**0.1879**
Bye-bye-2	**-2.8967**	**3.2323**	**-1.6911**	**0.3700**
Bye-bye-3	**-2.1305**	**1.8298**	**-1.0827**	**0.4333**
Bye-bye-4	**-1.8329**	**1.4566**	**-1.1583**	**0.6069**
Bye-bye-5	**-2.5383**	**2.6340**	**-1.4634**	**0.4105**
Bye-bye-6	**-1.6374**	**0.6464**	**-0.1703**	**0.1986**
Voice signal	Features with half sampling rate			
Bye-bye-1	**-2.8293**	**2.9235**	**-1.2684**	**0.1879**
Bye-bye-2	**-2.8967**	**3.2323**	**-1.6911**	**0.3700**
Bye-bye-3	**-2.1305**	**1.8298**	**-1.0827**	**0.4333**
Bye-bye-4	**-1.8329**	**1.4566**	**-1.1583**	**0.6069**
Bye-bye-5	**-2.5383**	**2.6340**	**-1.4634**	**0.4105**
Bye-bye-6	**-1.6374**	**0.6464**	**-0.1703**	**0.1986**
Voice signal	Features with doubling the amplitude			
Bye-bye-1	**-2.8293**	**2.9235**	**-1.2684**	**0.1879**
Bye-bye-2	**-2.8967**	**3.2323**	**-1.6911**	**0.3700**

Bye-bye-3	**-2.1305**	**1.8298**	**-1.0827**	**0.4333**
Bye-bye-4	**-1.8329**	**1.4566**	**-1.1583**	**0.6069**
Bye-bye-5	**-2.5383**	**2.6340**	**-1.4634**	**0.4105**
Bye-bye-6	**-1.6374**	**0.6464**	**-0.1703**	**0.1986**

From the results shown in table 2 we can see the following facts:

- The voice voiceprint (features) is stable and fixed.
- The voice voiceprint (features) is unique, thus we can use it to identify a person in a voice recognition system.
- Changing the voice sampling rate does not affect the voiceprint.
- Changing the voice amplitude does not affect the voiceprint.

Experiment 1: Using 4 order FIR to form voice features for the same different words spoken by the same person:
Table 3 shows the results of this experiment:

Table 3: Experiment 2 results

Voice signal	**Features(Original voice signal)**			
Bye-bye-1	**-2.8293**	**2.9235**	**-1.2684**	**0.1879**
Okay-1	**-2.7095**	**2.7998**	**-1.3446**	**0.2704**
Yes-1	**-1.9862**	**2.0603**	**-1.5439**	**0.4951**
No-1	**-2.7167**	**2.9875**	**-1.7150**	**0.4676**
Zero-1	**-1.5929**	**0.9468**	**-0.5265**	**0.2541**
Voice signal	Features with half sampling rate			
Bye-bye-1	**-2.8293**	**2.9235**	**-1.2684**	**0.1879**
Okay-1	**-2.7095**	**2.7998**	**-1.3446**	**0.2704**
Yes-1	**-1.9862**	**2.0603**	**-1.5439**	**0.4951**
No-1	**-2.7167**	**2.9875**	**-1.7150**	**0.4676**
Zero-1	**-1.5929**	**0.9468**	**-0.5265**	**0.2541**
Voice signal	**Features with doubling the amplitude**			
Bye-bye-1	**-2.8293**	**2.9235**	**-1.2684**	**0.1879**
Okay-1	**-2.7095**	**2.7998**	**-1.3446**	**0.2704**
Yes-1	**-1.9862**	**2.0603**	**-1.5439**	**0.4951**
No-1	**-2.7167**	**2.9875**	**-1.7150**	**0.4676**
Zero-1	**-1.5929**	**0.9468**	**-0.5265**	**0.2541**

From the results shown in table 3 we can see the following facts:

- The voice voiceprint (features) is stable and fixed.
- The voice voiceprint (features) is unique, thus we can use it to identify a spoken word by a human in a voice recognition system.
- Changing the voice sampling rate does not affect the voiceprint.
- Changing the voice amplitude does not affect the voiceprint.
- The voiceprint length is flexible and it can be easily changed by changing the order of FIR filter.

IV. CONCLUSION

LPC method of voice file features extraction was proposed, referring to the obtained experiment results we can conclude the following:

- LPC matlab function is suitable to create a voiceprint for each individual/word, this voiceprint is a unique for each individual/word and it can be easily used to recognize a person or a spoken word in voice recognition system.
- The extracted voiceprints are fixed for each voice file.
- The number of values within a voiceprint is flexible, and it can be changed by changing the order of the used FIR filter.

ISSN (Online) 2321-2004
ISSN (Print) 2321-5526

International Journal of Innovative Research in
Electrical, Electronics, Instrumentation and Control Engineering

Vol. 8, Issue 1, January 2020

REFERENCES

[1]. Jamil Al Azzeh, Hussein Alhatamleh, Ziad A Alqadi, Mohammad Khalil Abuzalata, Creating a Color Map to be used to Convert a Gray Image to Color Image, International Journal of Computer Applications, v. 975, pp. 8887, 2016.
[2]. Qazem Jaber Ziad Alqadi, Jamil azza, Statistical analysis of methods used to enhance color image histogram, XX International scientific and technical conference, 2017.
[3]. Jihad Nader, Ziad A. A. Alqadi, Bilal Zahran, Analysis of Color Image Filtering Methods, International Journal of Computer Applications, v. 174, issue 8, pp. 12-17, 2017.
[4]. J Al-Azzeh M Abuzalata, Ziad Alqadi, Modified Inverse LSB Method for Highly Secure Message Hiding, International Journal of Computer Science and Mobile Computing, v. 8, issue 2, pp. 93-103, 2019.
[5]. Majed O Al-Dwairi, Ziad A Alqadi, Amjad A Abujazar, Rushdi Abu Zneit, Optimized true-color image processing, World Applied Sciences Journal, v. 8, issue 10, pp. 1175-1182, 2010.
[6]. Jamil Al-Azzeh, Bilal Zahran, Ziad Alqadi, Belal Ayyoub, Muhammed Mesleh, A Novel Based On Image Blocking Method To Encrypt-Decrypt Color, International Journal on Informatics Visualization, v. 3, issue 1, pp. 86-93, 2019.
[7]. R. Szabo, A. Gontean, “Human Voice Signal Synthesis and Coding”, IFAC Proceedings Volumes, Vol. 46, No. 28, pp. 336-341, 2013.
[8]. K. M. Matrouk, A. Alhasanat, H. Alasha'ary, Z. Alqadi, H. M. AlShalabi, “Speech Fingerprint to Identify Isolated Word-Person”, World Applied Sciences Journal, Vol. 31, No. 10, pp. 1767-1771, 2014.
[9]. Jihad Nader Ismail Shayeb, Ziad Alqadi, Analysis of digital voice features extraction methods, International Journal of Educational Research and Development, v. 1, issue 4, pp. 49-55, 2019.
[10]. Majed O. Al-Dwairi, Amjad Y. Hendi, Mohamed S. Soliman, Ziad A.A. Alqadi, A new method for voice signal features creation, International Journal of Electrical and Computer Engineering (IJECE), v. 9, issue 5, pp. 4092-4098, 2019.
[11]. Ziad A. AlQadi Amjad Y. Hindi, Majed O. Dwairi, A Novel Technique for Data Steganography, Engineering, Technology & Applied Science Research, v. 9, issue 6, pp. 4942-4945, 2019.
[12]. Aws Al-Qaisi, Saleh A Khawatreh, Ahmad A Sharadqah, Ziad A Alqadi, Wave File Features Extraction Using Reduced LBP, International Journal of Electrical and Computer Engineering, v. 8, issue 5, pp. 2780, 2018.
[13]. Ruud M. Bolle, Jonathan H. Connell, Sharath Pankanti, Nalini K. Ratha, and Andrew W. Senior, Guide to Biometrics. Springer Science + Business Media, Inc, NY 10013, USA, 2004, pp 3 – 6, 31 – 45, 146 – 148.
[14]. (2010, August 10). Biometrics – Wikipedia, the free encyclopaedia [Online]. Available: http://en.wikipedia.org/wiki/Biometrics.
[15]. Tiwalade O. Majekodunmi, Francis E. Idachaba, A Review of the Fingerprint, Speaker Recognition, Face Recognition and Iris Recognition Based Biometric Identification Technologies, Proceedings of the World Congress on Engineering 2011 Vol II WCE 2011, July 6 - 8, 2011.
[16]. R. Szabo, A. Gontean, I. Lie, “Sound Based Coin Recognition and Clapper”, 16th International Conference on Soft Computing (MENDEL), pp. 509–516, June 23-25, 2010.
[17]. A. K. Paul, D. Das, M. M. Kamal, “Bangla Speech Recognition System Using LPC and ANN”, Seventh International Conference on Advances in Pattern Recognition (ICAPR), Kolkata, India, pp. 171– 174, February 4-9, 2009.
[18]. Ayman Al-Rawashdeh, Ziad Al-Qadi, Using wave equation to extract digital signal features, Engineering, Technology & Applied Science Research, v. 8, issue 4, pp. 1356-1359, 2018.
[19]. Saleh Khawatreh, Belal Ayyoub, Ashraf Abu-Ein, Ziad Alqadi, A Novel Methodology to Extract Voice Signal Features, International Journal of Computer Applications, v. 975, pp. 8887, 2018.
[20]. Ziad Alqadi, Bilal Zahran, Jihad Nader, Estimation and Tuning of FIR Low pass Digital Filter Parameters, international Journal of Advanced Research in Computer Science and Software Engineering, v. 7, issue 2, pp. 18-23, 2017.
[21]. Jihad Nader Ahmad Sharadqh , Ziad Al-Qadi, Bilal Zahran, Experimental Investigation of Wave File Compression-Decompression, International Journal of Computer Science and Information Security, v. 14, issue 10, pp. 774-780, 2016.
[22]. Ashraf Abu-Ein, Ziad AA Alqadi, Jihad Nader, A Technique of Hiding Secretes Text in Wave File, International Journal of Computer Applications, 2016.
[23]. Jihad Nadir, Ashraf Abu Ein, Ziad Alqadi, A Technique to Encrypt-decrypt Stereo Wave File, International Journal of Computer and Information Technology, v. 5, issue 5, pp. 465-470, 2016.
[24]. A Waheeb, Ziad AlQadi, Gray image reconstruction, Eur. J. Sci. Res, v. 27, pp. 167-173, 2007.
[25]. Jamil Azzeh, Bilal Zahran, Ziad Alqadi, Salt and Pepper Noise: Effects and Removal, International Journal on Informatics Visualization, v. 2, issue 4, pp. 252-256, 2018.
[26]. ZIAD ALQADI, A Modified LBP Method to Extract Features from Color Images, Journal of Theoretical and Applied Information Technology, v. 96, issue 10, pp. 3014-3024, 2018.
[27]. Bilal Zahran Belal Ayyoub, Jihad Nader, Ziad Al-Qadi, Suggested Method to Create Color Image Features Victor`, Journal of Engineering and Applied Sciences, v. 14, issue 1, pp. 2203-2207, 2019.
[28]. Dr. Ziad Alqadi, Akram Mustafa, Majed Alduari, Rushdi Abu Zneit, True color image enhancement using morphological operations, International review on computer and software, v. 4, issue 5, pp. 557-562, 2009.
[29]. Ziad AlQadi, Hussein M Elsayyed, Window Averaging Method to Create a Feature Victor for RGB Color Image, International Journal of Computer Science and Mobile Computing, v. 6, issue 2, pp. 60-66, 2017.
[30]. R Abu Zneit, Ziad AlQadi, M Abu Zalata, A Methodology to Create a Fingerprint for RGB Color Image, International Journal of Computer Science and Mobile Computing, v. 16, issue 1, pp. 205-212, 2017.
[31]. Mohammed Ashraf Al Zudool, Saleh Khawatreh, Ziad A. Alqadi, Efficient Methods used to Extract Color Image Features, IJCSMC, v. 6, issue 12, pp. 7-14, 2017.
[32]. Ahmad Sharadqh Naseem Asad, Ismail Shayeb, Qazem Jaber, Belal Ayyoub, Ziad Alqadi, Creating a Stable and Fixed Features Array for Digital Color Image, IJCSMC, v. 8, issue 8, pp. 50-56, 2019.
[33]. Ahmad Sharadqh Jamil Al-Azzeh , Rashad Rasras , Ziad Alqadi , Belal Ayyoub, Adaptation of matlab K-means clustering function to create Color Image Features, International Journal of Research in Advanced Engineering and Technology, v. 5, issue 2, pp. 10-18, 2019.
[34]. Yousif El-tous Dr. Ghazi M. Qaryouti, Mohammad Abuzalata, Ziad Al Qadi, A METHOD TO CREATE VOICE SIGNAL HISTOGRAM, IJRSR, vol. 10, issue 12, pp. 36650-36654, 2020.
[35]. Ziad Alqadi, Prof. Yousf Eltous Dr. Ghazi M. Qaryouti, Prof. Mohammad Abuzalata, Analysis of Digital Signal Features Extraction Based on LBP Operator, International Journal of Advanced Research in Computer and Communication Engineering, vol. 9, issue 1, pp. 1-7, 2020.
[36]. Prof. Yousif Eltous, Dr. Ghazi M. Qaryouti, Prof. Mohammad Abuzalata, Prof. Ziad Alqadi, Evaluation of Fuzzy and C_mean Clustering Methods used to Generate Voiceprint, IJCSMC, vol. 9, issue 1, pp. 75 -83, 2020.